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INSTRUCTION MANUAL

FOR MODEL 2A/2B LINE SIMULATOR

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(Revised)

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OPERATING NOTICE

The following list anticipates the most common operating errors.

1. The blue 7-pin socket in the upper left corner of the panel must have the 2B connector or a jumper connector in place. The signal path passes through this connector.

2. For serial numbers below 400, when operating with an external input the peak to peak input level should not exceed 2.2 volts. This corresponds to 0 dBm for a sine wave but it corresponds to perhaps -8 dBm for a QAM modem.

3. The SFI dial *adds* a sinusoid at the modulation frequency to the output. If this frequency is below 30 Hz, it will be attenuated by the output transformer. In this case it is not the SFI dial which is "out of calibration," the problem is loss in the output transformer and this is an invalid operating mode.

4. When operating with a external input, the SFI calibration is relative to 0 dBm input - see Section 5-6.

5. The RISE TIME, HIT DURATION, and INTERVAL hit controls can be set (incorrectly) for anomalous results. To avoid confusion the following conditions must be observed:
(1) HIT DURATION + HIT RISE TIME shorter than INTERVAL
(2) HIT DURATION longer than HIT RISE TIME

6. When the 2B unit is used ahead of a steady state line simulator, e.g., 3000, C1, C2, etc., the frequency response of the line simulator modifies the dial calibration of the harmonic generator.

7. The noise generator spectrum is not C-message weighted. It is flat. In the real world C-message weighting is not introduced by the facility, it is introduced by the test instrument. If desired the panel dials can be scaled, by an internal adjustment, so as to agree with a C-message calibrated noise meter. However, this does *not* affect the noise spectrum.

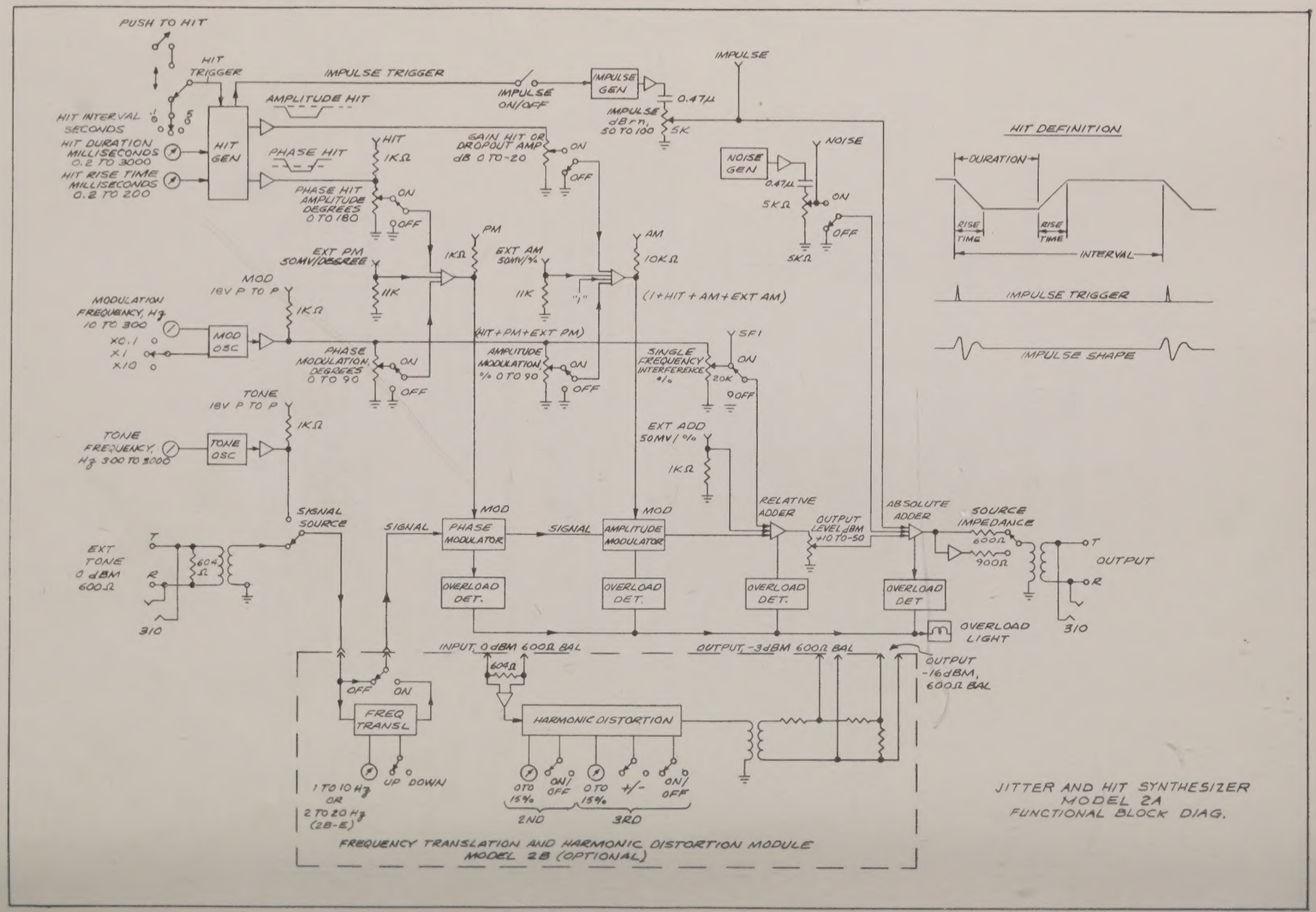


Figure 1 2A/2B Block Diagram

SECTION 1. OPERATING INSTRUCTIONS

1. INTRODUCTION

The 2A Synthesizer, without the 2B option, is furnished with a jumper connector in place. With the 2B option, the cable to the 2B is in place. One or the other of these must be plugged into the 7 pin blue receptacle in the upper left corner of the 2A panel, otherwise the signal path is interrupted.

Referring to block diagram, figure 1, the EXT TONE signal or the internal oscillator (as selected by the SIGNAL SOURCE switch) passes through a frequency translator (optional), the phase modulator, an amplitude modulator, an SFI (Single Frequency Interference) Adder, a calibrated attenuator, and a Steady State and Impulse Noise Adder. The Tone Oscillator permits the generation of test tones with calibrated impairments for evaluation and/or calibration of jitter, noise, and hit sets, as well as for the training of maintenance personnel. The Modulation Oscillator provides a second variable frequency source which may be applied in calibrated amplitude to the Phase Modulator to generate phase jitter, to the Amplitude Modulator to generate amplitude jitter, and/or to the SFI adder to generate single frequency interference. Access is provided for the application of external inputs of any desired (voice band) spectral content to the Phase and Amplitude Modulators and the SFI Adder. The Hit Generator provides Trapezoidal shaped pulses of adjustable Rise Time, and adjustable Duration. These pulses may be manually initiated or repetitively generated at a selectable Interval. These pulses may be selectively applied to generate Phase Hits, Gain Hits, Coincident Phase and Gain Hits, or Dropouts. The modulation-impaired signal is then coupled, via a calibrated attenuator, to the Steady and Impulse Noise Adder where these latter two impairments may be added. The Impulse Generator is triggered by the leading edge of the Hit Generator pulse. The impulse has an approximately uniform spectral energy distribution from 800 to 2200 Hz.

The harmonic distortion section of the 2B module is patchable at both input and output so that it can be placed ahead of or behind the 2A in the signal path.

2. 2A CONTROL FUNCTIONS (Refer to front panel photo of instrument, figure 2) The heavy arrows starting at the lower left side of the panel and ending at the OUTPUT jack (lower right) define the signal flow path. Panel controls and indications perform the functions described below:

a. SIGNAL SOURCE – Selects EXT TONE jack or internal tone oscillator as signal source.

b. Tone Controls

TONE FREQUENCY, HZ – Adjusts internal tone oscillator frequency from 300 to 3000 Hz.

OUTPUT LEVEL selector switch and vernier - Adjusts output level between -50 and +10 dBm. Add vernier reading to switch position for reading. The output level attenuator is direct reading for an EXT TONE input of 0 level and 600 ohms source impedance. The EXT TONE input *must* be less than 2.2 volts peak to peak. This is the peak to peak value of a 0 dBm (600 ohms) sine wave.

SOURCE IMPEDANCE switch – Selects source impedance of balanced output as either 600 or 900 ohms.

c. Modulation Controls

MODULATION FREQUENCY, HZ – Range switch and vernier, adjusts modulation from 1 to 3000 Hz.

PHASE MODULATION, DEGREES – Vernier adjustment and on/off switch, adjusts amplitude of phase modulation from 0 to 90° peak to peak. Phase modulation is at MODULATION FREQUENCY.

EXT PM jack – Provides external access to the phase modulator input. This input will accommodate voice band inputs of any spectral content and phase modulate the signal with them. Scale factor is nominally 50 millivolts per degree.

AMPLITUDE MODULATION, % – Vernier adjustment and on/off switch, adjusts amplitude of AM from 0 to 90%. AM is at MODULATION FREQUENCY.

EXT AM jack – Provides external access to the amplitude modulator input. This input will accommodate voice band inputs of any spectral content and amplitude modulate the signal with them. Scale factor is nominally 50 millivolts per percent.

SINGLE FREQUENCY INTERFERENCE % – Vernier adjustment and on/off switch adjusts amplitude of additive tone from 0 to 90% of tone source amplitude. Additive tone is at MODULATION FREQUENCY.

EXT ADD jack – Provides external access to the SFI adder. This input will accommodate voice band inputs of any spectral content and add them to the output. Scale factor is nominally 50 millivolts per percent.

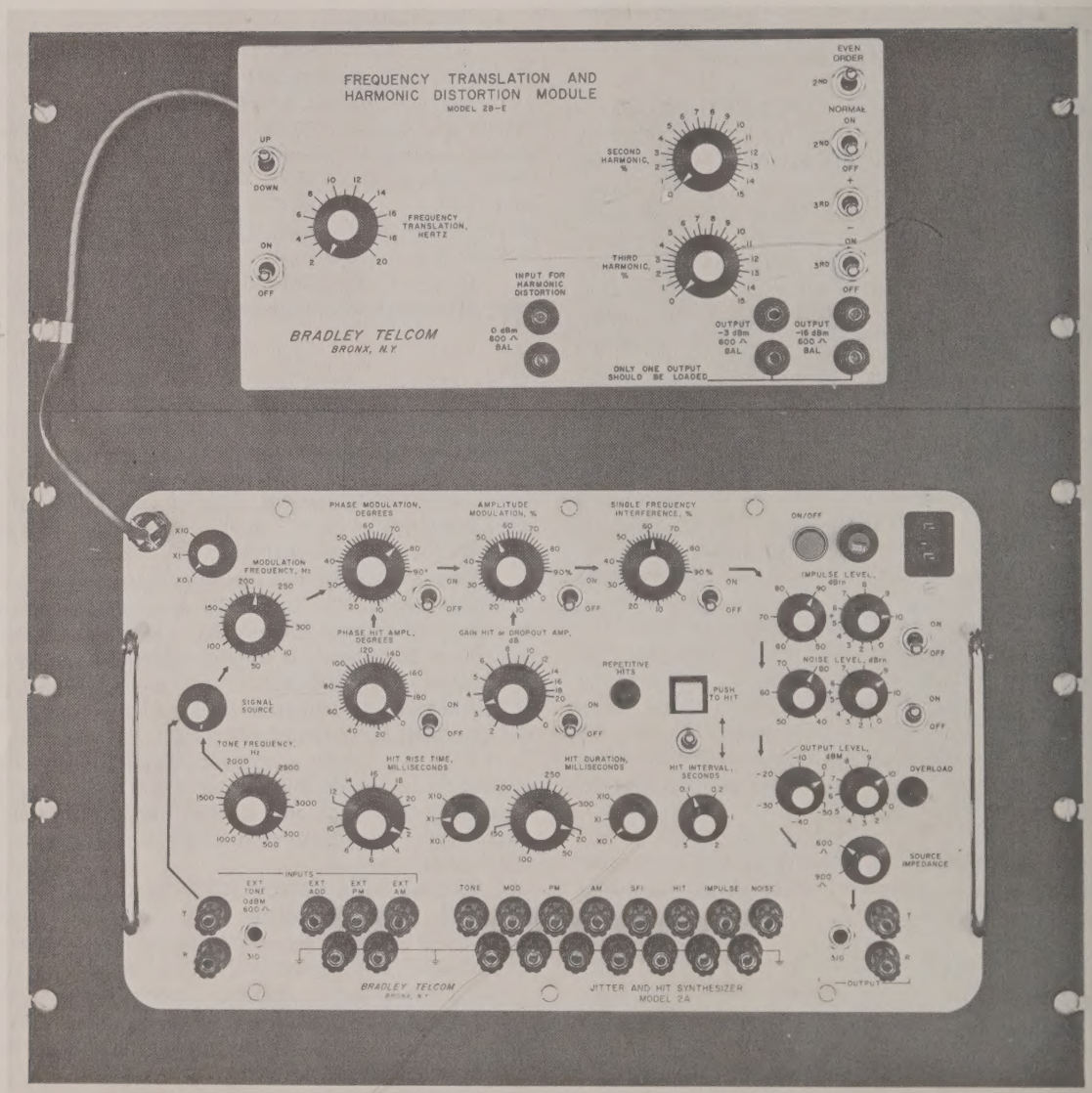


Figure 2 2A/2B Rack Mounting Unit

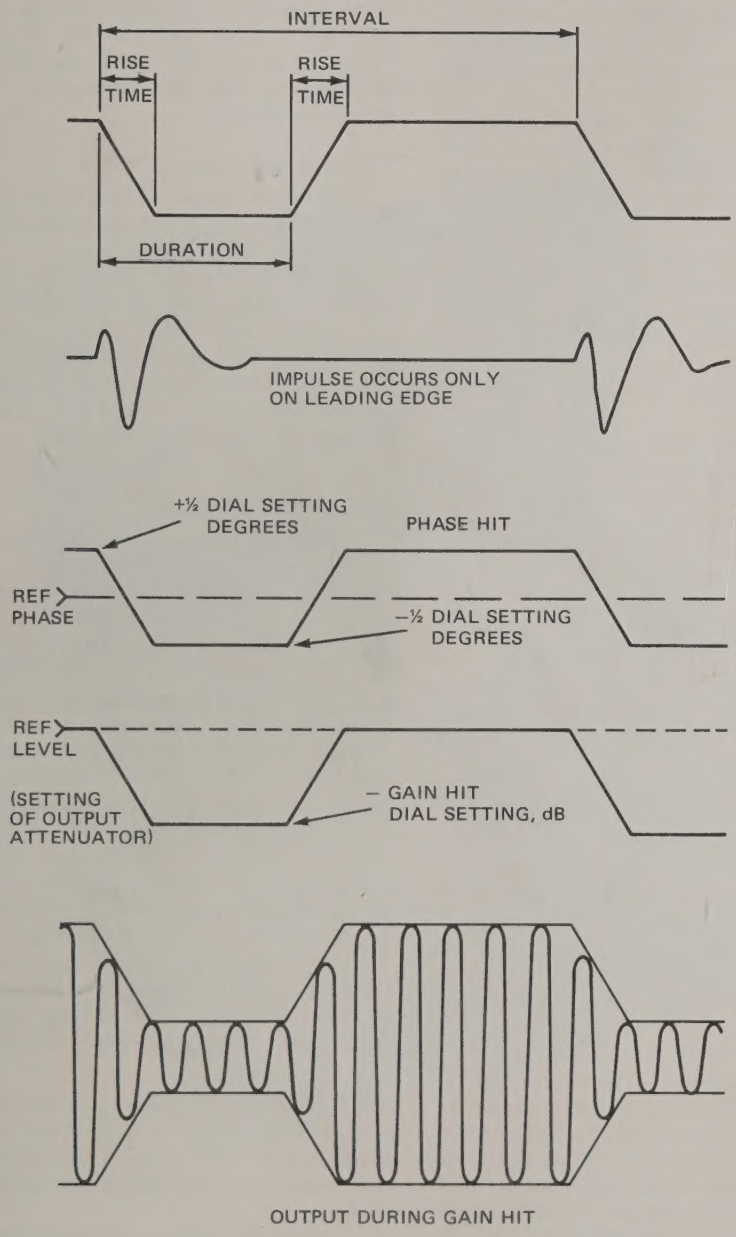


Figure 3 Hit Definitions

d. Additive Disturbances

IMPULSE LEVEL – Selector switch, vernier, and on/off switch. This set of controls adds a calibrated impulse to the OUTPUT. Note all hits other than impulse occur in pairs, see below, the impulse coincides only with the first hit of each pair. Range is from 50 to 100 dBrn. This is instantaneous power at the peak of the impulse. Add switch and vernier setting for reading.

NOISE LEVEL – Selector switch, vernier, and on/off switch. This set of controls adds a calibrated amount of white noise (3KC band width) to the OUTPUT. Range is from 40 to 90 dBrn. Add switch and vernier setting for reading.

e. Hit Disturbances, see Figure 3

Method of hit initiation is selected by an "arrow labled" toggle switch between the HIT INTERVAL switch and the PUSH TO HIT button. In the "down" position of the toggle, the HIT INTERVAL Switch controls, and in the "up" position the PUSH TO HIT button controls.

HIT INTERVAL, SECONDS – Selects one of the five designated repetitive hit periods.

PUSH TO HIT – momentary push button - This switch causes a gain or phase hit on depression and another hit in the opposite direction upon release. If the button is released prior to the expiration of the HIT DURATION, the second hit will not occur until the end of the HIT DURATION interval. Impulse hits occur only upon depression of the button and not upon release.

HIT DURATION, MILLISECONDS – Range switch and vernier, adjusts hit duration from 20 to 3000 milliseconds. Multiply scale switch setting and vernier reading for reading. Note HIT DURATION can be made longer than HIT INTERVAL. This will cause seemingly erratic hit intervals. It is not a normal mode of operation.

HIT RISE TIME, MILLISECONDS – Scale switch and vernier, adjusts hit rise time from 0.2 to 200 milliseconds. Multiply scale switch setting and vernier for reading. Note HIT RISE TIME can be set longer than HIT DURATION and HIT INTERVAL. This will result in seemingly erratic and/or reduced hit amplitude. It is not a normal mode of operation. To avoid confusion the following conditions must be observed:

(1) HIT DURATION + HIT RISE TIME shorter than INTERVAL

(2) HIT DURATION longer than HIT RISE TIME

PHASE HIT AMPL, DEGREES – Vernier and on/off switch adjusts the amplitude of the phase hit. PHASE HIT AMPL vernier offset the output phase by half of its set value in the lagging direction. Then when the hit occurs, the phase moves to an equal amount (half its set value) in the lead direction at the start of the HIT INTERVAL. This occurs with a linear ramp whose duration is set by the HIT RISE TIME. At the expiration of the HIT DURATION the phase returns to the original lagging direction, again with the same linear phase ramp.

GAIN HIT AMPL, DEGREES – Vernier and on/off switch adjusts the amplitude of the gain hit. When the hits occur, the level drops by a dB amount as indicated by the GAIN HIT dial setting. The drop occurs with a linear ramp whose duration is set by the HIT RISE TIME. At the expiration of the HIT DURATION, the level returns to its starting value with the same linear ramp. Note that the gain hit does not affect the noise, SFI, or EXT ADD components. These are added to the signal after the gain hit is introduced, see block diagram. The gain hit will affect noise which is introduced at the EXT TONE jack when that input is being processed by the synthesizer, i.e., TONE SOURCE in appropriate position.

f. Indicator Lights

ON/OFF (green) – This is an alternate action on/off switch which controls the application of primary power. Line power is fused at 1 ampere.

REPETITIVE HITS (green) – This indicates that the HIT INTERVAL switch is in control of hit operation.

OVERLOAD (red) – This light indicates that the combination of impairments called for by the control settings and/or the external inputs may be causing the instrument to saturate, and that the output is not a valid representation of the input plus specified impairments.

g. Control Outputs

TONE (jack) – Nominally 18 volts peak to peak sinusoid at TONE FREQUENCY, 1000 ohm source impedance. This output is useful for oscilloscope syncing and as a general purpose oscillator.

MOD (jack) – Nominally 18 volts peak to peak sinusoid at MODULATION FREQUENCY, 1000 ohm source impedance. Useful as a general purpose oscillator, and for scope syncing to display modulation.

PM (jack) – Summation of internal phase modulation, phase hit, and EXT PM. Scale factor nominally 50 mv per degree, source impedance 1000 ohms. This signal is the phase modulation forcing function which exists at any given time.

AM (jack) – Summation of internal amplitude modulation, gain hit, and EXT AM. Scale factor nominally 50 mv %, source impedance 10K. This signal is the amplitude modulation forcing function which exists at any given time.

SFI (jack) – SFI setting. Scale factor nominally 50 per mv per %, source impedance 0 to 5000 ohms. This signal exists at the jack regardless of the state of the SINGLE FREQUENCY INTERFERENCE ON/OFF switch.

HIT (jack) – This is the hit profile for oscilloscope display, the voltage swing is from +4.5 to -4.5V, source impedance is 1000 ohms. This voltage may be used to synchronize an oscilloscope for stationary display of hit profile.

IMPULSE (jack) – Output of impulse generator, 1000 to 5000 ohms source impedance. This voltage may be displayed on an oscilloscope. It is only present when the IMPULSE on/off switch is ON.

NOISE (jack) – Output of the noise generator independent of the state of the NOISE LEVEL ON/OFF switch. May be used as a general purpose noise source. 1000 to 5000 ohms source impedance.

Figure 4 2A/2B Portable Unit

3. 2B CONTROL FUNCTIONS. The 2B Frequency Translation and Harmonic Distortion Module mounts inside the lid of the Model 2A Jitter and Hit Synthesizer, see figure 4. It receives power (+12, -12, and ground) and signal from the 2A via the connector on the upper left panel of the 2A. It also returns the frequency translated signal via this connector. The input and output for harmonic distortion are separately patchable on the panel of the 2B to permit insertion of harmonic distortion ahead of or behind the impairments inserted by the basic 2A unit.

Panel elements of the 2B are as follows:

FREQUENCY TRANSLATION, HERTZ – Calibrated continuous adjustment from 1 to 10 Hertz of translating frequency.

UP/DOWN – Two position toggle, selects direction of frequency translator.

ON/OFF – Two position toggle, inserts frequency translation in signal path or bypasses frequency translator.

The frequency translation mechanization in the 2B introduces 2 to 4% of amplitude modulation and 3 or 4 degrees of phase jitter at the translation frequency. The UP/DOWN switch may be used to insert frequency hits, e.g., set to 2 Hz, switch from UP to DOWN to put in a -4 Hz frequency hit.

INPUT FOR HARMONIC DISTORTION – 0 dBm, 600 ohms, Balanced-Sinusoid should be at 0 dBm level here for accurate calibration of the THIRD HARMONIC dial and of the SECOND HARMONIC dial for the NORMAL position of the NORMAL/EVEN ORDER switch. In the EVEN ORDER position of this switch the SECOND HARMONIC dial calibration is not input level sensitive.

SECOND HARMONIC % vernier, NORMAL/EVEN ORDER toggle, ON/OFF Toggle Controls operation and level of second harmonic or even order distortion, 0 to 15%.

EVEN ORDER/NORMAL – Selects harmonic mode – NORMAL – Output includes pure 2nd, see 4. below, EVEN ORDER – Output includes Dial Set x (2nd H - 1/5 4th H + 1/12th 6th H - 1/30 8th H etc.)

THIRD HARMONIC, % vernier and ON/OFF Toggle – Controls operation and level of third harmonic distortion, 0 to 15%.

THIRD HARMONIC +/ – Controls sense of third harmonic.

Output, -3 dBm, 600 ohms balanced -

and OUTPUT, -16 dBm, 600 ohms balanced -

Only one of these outputs should be used at any time.

4. HARMONIC DISTORTION

The harmonic distortion function is patched from the panel of the 2B so that it can be inserted either ahead of or behind the other 2A impairments. The performance equation for this unit is:

$$\text{Output} = \Big\{ \Sigma A_i \cos w_i t + \text{Dial Set 2nd} \times \frac{2}{A_o^2} \left(\Sigma A_i^2 \cos 2 w_i t - \Sigma A_i^2 \right) + \text{Dial Set 3rd} \times \frac{2}{A_o^2} \Big[(\Sigma A_i \cos w_i t) \left(\Sigma A_i^2 \cos 2 w_i t - \Sigma A_i^2 \right) - \Sigma A_i^3 \cos w_i t \Big] \Big\} - 3 \text{ dB or} - 16 \text{ dB}$$

Where Ao = coefficient of 0 dBm sine wave, i.e., Ao cos wt = "0" level Ai cos wt are components of complex wave.

The calibration of the dials is dependent on the level and frequency distribution of the input signal, and is mathematically predictable from the above.

For a sinusoidal input at 0 level, i.e.,

Output voltage = (Ao cos wt + Dial Set 2nd x cos 2 wt + Dial Set 3rd x cos 3 wt) –3 dB or –16 dB depending upon the output jack selected.

SECTION 2 – ALIGNMENT PROCEDURE SYNTHESIZER MODEL 2A/2B

1. INTRODUCTION

Measurements indicating the need for recalibration should be double and triple checked for control setting and/or measurement errors before undertaking readjustment of the Synthesizer. A systematic procedure is given below.

CAUTION

There are individual potentiometers which independently affect the scale factor of each dial function. These may be adjusted and will not interact with any other adjustment. There are also adjustments which affect the quality of operation and which interact. A list is given below of the adjustments which are independent. To adjust any single impairment scaling, the appropriate potentiometer should be located and adjusted. Unless one desires to go through the entire systematic procedure also given further below, adjustment of any other potentiometers should be avoided.

To align the synthesizer it is necessary to have an extender card, a dual trace oscilloscope, a voice band level meter, a digital volt meter and a frequency counter capable of resolving 1/10 Hz up to 3000 Hz. The card locations are identified in Figure 5, the adjustment potentiometer locations on individual cards are indicated in Figure 6. The 2B adjustment locations are given in Figure 8. In many cases it is necessary to use the extender card to reach adjustments.

Note the signal path is interrupted at the 7 pin connector at the upper left corner of the front panel. Either the 2B frequency translation and harmonic distortion module connector should be plugged in here, or a jumper connector which connects pin D to E and pin A to B. Caution, pins F, C, and H are +12, -12 and ground. Do not short these to each other or to the other pins.

Mechanical dissassembly of the 2A is accomplished by removing six 10-32 screws, three each along the top edge and bottom edge of the panel. The unit may then be lifted out of the case. Adjustments can be made on the Phase Splitter and Phase Modulator in this condition also a few of the other adjustments may be reached with a long thin bladed screwdriver.

Access to the adjustments on the Oscillator, Control Unit, Amplitude Modulator, and Output Amplifier will require that the 2A rear cover be removed and that these cards be placed on an extender card. The extender cards do not have keying slots cut out, as do the individual cards. It will be necessary to remove the plastic key tab in the female connector using long nose pliers. Be careful in doing so not to damage the connector pins. When the adjustment procedure is finished replace the key tab. The tab not only prevents wrong insertion of the card, but it guides the card connector relative to the female connector. If anomalous behavior is encountered while a card is on an extender, make sure that the problem is not in the connector by jiggling, removal and reinsertion, or by checking continuity through with an ohm meter.

To reach the 2B adjustments, remove six 6-32 screws holding the 2B in the cover.

CAUTION

It is necessary that the 2B connector be plugged into the 2A to accomplish any adjustments involving the signal path.

2. INDIVIDUAL SCALING ADJUSTMENT POTENTIOMETERS

Gain from "Ext Tone" to Output — R68 of Output Amplifier

Place SIGNAL SOURCE switch in EXT TONE position, OUTPUT LEVEL switch to -10, and OUTPUT LEVEL vernier to +10. Patch 1000 Hz sine wave at "O" level or lower to EXT TONE, and observe OUTPUT level to be the same as the input level.

Output Level of Internal Tone Oscillator — R82 of Amplitude Modulator

Place SIGNAL SOURCE switch in TONE FREQUENCY position, set TONE FREQUENCY TO 1000 Hz, and OUTPUT LEVEL attenuator to +10, i.e., switch to "O" vernier to +10 and SOURCE IMPEDANCE to 600 ohms. Observe that output level is +10. Note output must be loaded by 600 ohms.

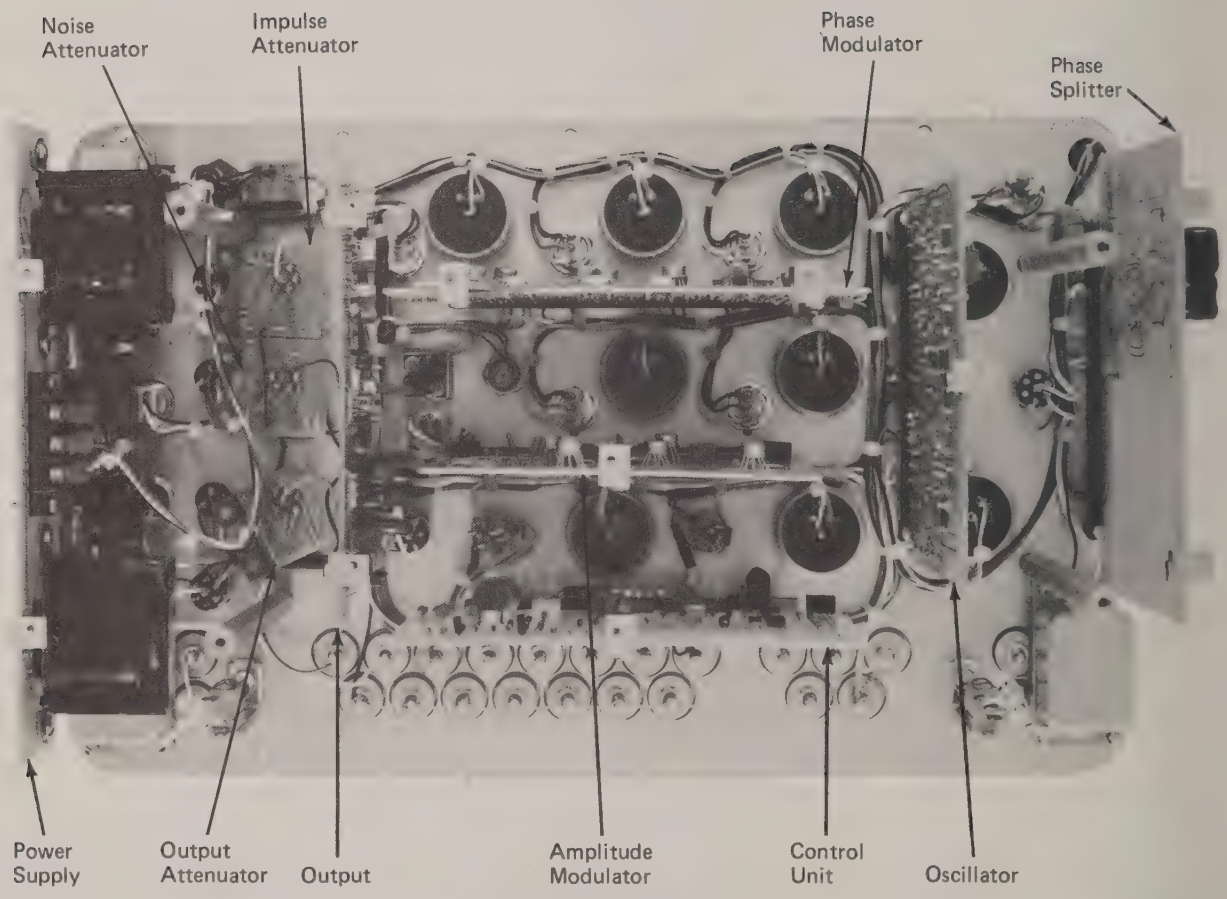
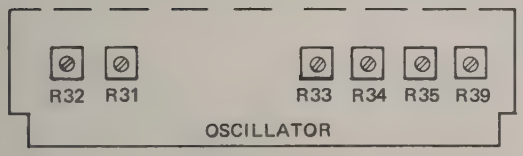


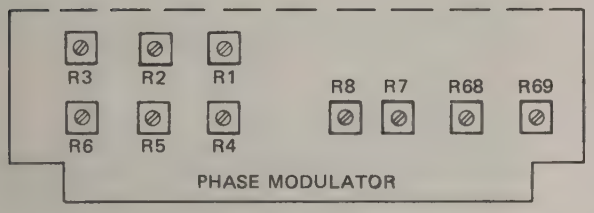
Figure 5 2A Rear Plate Removed

Figure 5 2A, Rear Plate Removed

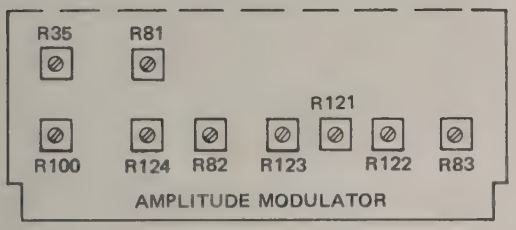
R1 Balance
R2 Cos wt Level
R3 Sin wt Level
R4 Relative level, Modem to internal oscillator



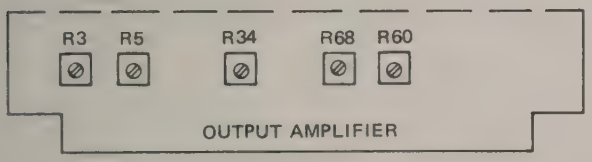
R31 Tone duty cycle
R32 Tone Frequency
R39 Modulation duty cycle
R33 Modulation frequency, x0.1 range
R34 Modulation frequency, x 1 range
R35 Modulation frequency, x10 range



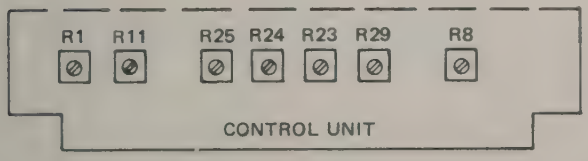
R1 Cos wt sin mt dc zero
R2 Sin mt zero
R3 Cos wt zero
R4 Sin wt cos mt dc zero
R5 Cos mt zero
R6 Sin wt zero
R7 Cos wt ±balance
R8 Cos wt peak value
R68 Phase hit scale
R69 Phase modulation scale



R35 Minimum duration adj
R100 Gain Hit balance
R81 Gain hit scale
R124 AM scale
R82 Output level
R121 Modulation zero
R122 Tone zero
R123 AM output balance
R83 PM sin/cos balance



R3 Impulse shape
R5 Impulse amplitude
R34 Noise level
R60 SFI scale
R68 Signal gain



R1 Rise time/Fall time symmetry
R11 Min rise time
R25 x 10 Duration adj
R24 x 1 Duration adj
R23 x 0.1 Duration adj
R29 Interval adj
R8 Rise time adj

Figure 6 2A, Adjustment Locations

SFI Scaling R60 of Output Amplifier

With No external input, the input selector is the EXT TONE position, all other impairments off and OUTPUT LEVEL to +10 dBm. Turn SFI on and adjust vernier for 10%. Read level on OUTPUT and adjust R60 so that reading is 20 dB below OUTPUT LEVEL dial reading, i.e., -10 dBm reading of level meter.

Phase Modulation Scaling R69 of Phase Modulator

Set in 90° PM, sync scope on TONE. Set scope on a high gain setting (so that zero crossings are distinct) check zero of scope carefully. Adjust modulation frequency for relatively high beat frequency between tone and modulation frequency so that pattern is easily observed. Adjust R69 for equal open and sweep traced interval.

Phase Hit Scaling R68 Phase Modulator

Set in 90° phase hits, 10 per second, 50 milliseconds duration, 0.2 millisecond rise time. Sync DC coupled scope on TONE and observe 90° hits, using the same techniques as for PM above. R68 adjusts this.

Amplitude Modulation Scaling R124 Amplitude Modulator

Set in some convenient AM level such as 50%, sync DC coupled scope on TONE and observe 50% AM while switching AM on and off. R124 adjusts this.

Gain Hit Zero R100 Amplitude Modulator

Switch GAIN HIT or DROPOUT with no repetitive hits, toggle in up position. Move GAIN HIT pot from 0 to -20 and observe no change in output level. R100 adjusts this.

Gain Hit Scaling R81 Amplitude Modulator

Set OUTPUT LEVEL to some convenient value say "O" dBm and GAIN HIT dial to -20. Depress PUSH TO HIT button and observe 20dB change in level. R81 adjusts this.

Impulse Scaling R5 Output Amplifier

With no input to EXT TONE place SIGNAL SOURCE switch in EXT TONE position. Turn IMPULSE on and set for repetitive hits at 0.2 second interval, sync scope on impulse and observe amplitude of highest (second) peak. With IMPULSE LEVEL set for 100dBRn this should be 2.45 volts. R5 adjusts this.

Noise Scaling R34, Output Amplifier

Turn off impulse above and turn on Noise. Set NOISE LEVEL to 90 dBRn and, using whatever weighting is desired, measure noise level. R34 adjusts this.
Note internal noise in bandwidth limited (3.5KHz flat) and the unit is shipped with this adjusted for correct reading, using a true rms level meter.

Second Harmonic Scaling R98, 2B

Patch OdBm, 1000 Hz, 600 ohms source sinusoid to INPUT FOR HARMONIC DISTORTION. Observe either -3 or -16 dBm 2B OUTPUT. Set SECOND HARMONIC dial to 10%, NORMAL/EVEN ORDER switch in NORMAL. Switch second harmonic off and on while observing output with PB-1 or PB-1C in the second harmonic mode or with a spectrum analyzer. Observe a -20 dB change in separation between the fundamental and the second harmonic as the switch is operated. This means that the residual second harmonic in the tone with the 2B second harmonic off is not counted as part of the -20 dB separation step when the harmonic is switched on.

CAUTION

This adjustment affects the calibration of the THIRD HARMONIC dial. Thus, if R98 is adjusted as above, the R111, the third harmonic scaling adjustment must be readjusted.

Third Harmonic Scaling R111, 2B

With same connection and input as for second harmonic adjustment above, and the THIRD HARMONIC dial set for 10%, observe a -20 dB separation step as the harmonic is switched on and off and a -10 dB separation step as the harmonic, when on, is switched from plus to minus.

2B Output Level Scaling R129, 2B

With "O" dBm input observe that the OUTPUT, 3dB level is -3dBm.

3. COMPLETE ADJUSTMENT SEQUENCE

Many of the following adjustments interact. The following breaks the adjustments down to sequences of independent groups and indicates the order in which they should be done.

a. PROCEDURE 1 – POWER SUPPLY

1. Adjust the potentiometer for 12 volt (+ and -) output. -12 exists on pins 1 through 7 of the power supply card and connector, plus 12 exists on pins 18 through 24, and ground exists on pins 8 through 13.

2. Observe that the power supply is balanced to within 10 millivolts. This balance is controlled by the equality of two 10K metal firm resistors which are located, together with a ceramic capacitor, between the two M301A metal can op amps in the center of the card toward the connector end. If the power supply balance is out, establish equality by shunting one or the other of the resistors to determine the change that must be made. Replace or shunt the appropriate resistors.

b. PROCEDURE 2 – OSCILLATOR ALIGNMENT

TONE NOTE:
The Tone Frequency is available at the TONE output jack on the panel, and this is a convenient place to plug in a frequency counter. The unloaded level at this point is 18 volts peak to peak.

1. Set R31 for exactly 50% duty cycle of the tone gate, which is available on pin 16 of the oscillator card. This can also be done with the Bradley Telcom PB series VF Line Analyzer (at 1000 Hz) or with a spectrum analyzer by adjusting R31 for minimum second harmonic at the TONE jack.

2. Phase the TONE FREQ dial for 300 Hz oscillator frequency at 300 Hz dial reading.

3. Rotate the dial to 3000 Hz and adjust R32 for 3000 Hz Oscillator frequency.

4. Check the frequency at 1500 Hz dial setting. It should be between 1450 and 1550 Hz.

MODULATION

The modulation frequency is available at the MOD jack in the front panel. This is a convenient place to plug in a frequency counter. The unloaded level at this point is nominally 18 volts peak to peak.

1. Set R39 for exactly 50% duty cycle of the modulation gate which is available on pin 15 of the card. This can also be done with one of the Bradley Telcom PB series VF Line Analyzers (at 1000 Hz) or with a spectrum analyzer observing the Modulation output (at 1000 Hz for the PB-1) and minimizing the second harmonic content.

2. With the modulation range switch in the x10 position, phase the MODULATION FREQ dial to 100 Hz modulation frequency at 10 Hz dial reading.

3. Rotate the dial to 300 Hz and adjust R35 for 3000 Hz frequency.

4. Set the range switch to x1 position and adjust R34 for 300 Hz modulation frequency.

5. Set the range switch to x0.1 position and adjust R33 for 30 Hz modulation frequency.

6. Check the modulation frequencies on all three ranges at approximate dial midpoint. The frequency should be within half a division accuracy.

c. PROCEDURE 3 – HIT CONTROLS

This procedure involves setting the potentiometers on the Control Unit card as well as one potentiometer on the Amplitude Modulator. During this procedure, the Amplitude Modulator card must be plugged in. For convenience in adjustment the Phase Modulator card may be removed and the Control Unit card placed on an extender. This permits access to all the relevant adjustments at the same time.

1. Set the hit selector toggle switch "down", the HIT INTERVAL switch to 0.1 second, observe pin 15 of the Control Unit connector or the HIT output jack with an oscilloscope self-sync, and adjust R29 of the Control Unit for 0.1 second pulse spacing.

2. Check 0.2, 1, 2, and 5 positions of the HIT INTERVAL switch, and trim R29 for best compromise on all five pulse spacings.

3. Phase the HIT DURATION dial against CCW stop at a dial reading of 20. Place the HIT DURATION range switch in the x0.1 position, and then adjust R35 of the Amplitude Modulator card for 2 milliseconds' duration of the pulse observed at pin 22 of the Control Unit. This pulse occurs at an interval which is controlled by the HIT INTERVAL switch and its duration is controlled by the setting of R35 of the Amplitude Modulator card.

4. Rotate the HIT DURATION dial to 300 and adjust R23 of the Control Unit for 30 millisecond duration. Recheck the dial at a reading of 20 milliseconds, and readjust R35 if necessary.

5. Set the HIT DURATION range switch to x1, and the dial to 300. Adjust R24 of the Control Unit to 300 milliseconds duration. This will require that the HIT INTERVAL be in the one-second, two-second, or five-second position, otherwise hit triggers will occur before the duration pulse is finished (and will be ignored).

6. Set the HIT DURATION range switch to x10 and adjust R25 for three seconds' duration. This will require that the HIT INTERVAL switch be in the five-second position.

7. Check mid-range for all three durations, and adjust and retouch the adjustments of all four potentiometers for best fit.

8. Observe pin 10 of the Control Unit connector while syncing the oscilloscope from pin 15 or from the HIT output jack.

9. Set the RISE TIME dial at its lower stop, reading of 2, the RISE TIME range switch at x0.1. Adjust R11 of the Control Unit for 0.2 milliseconds rise time.

10. Set the RISE TIME dial to 20 and adjust R8 of the Control Unit for 2 milliseconds rise time.

11. Check with the HIT RISE TIME range switch in the x1 and x10 positions for 20 and 200 milliseconds at the high end, and 2 and 20 milliseconds at the low end of the dial. Reset R11 and R8 for best fit all ranges. (middle of the range is most important).

d. PROCEDURE 4 – OUTPUT AMPLIFIER (NOISE AND IMPULSE)

1. With a terminated level meter on the output, the tone OFF, the SOURCE IMPEDANCE set for 600 ohms and the noise ON, adjust R34 so that the level meter reads the noise dial setting when the noise dial is set for 90 dBrn; that is with the noise level attenuator switch in the 80 position and the vernier set with a reading of +10. 90 dBrn is .774 volts rms. NOTE: That if an average reading calibrated rms level meter is used, it will read 1dB low for white noise.

2. Impulse Shape. Turn the impulse on and observe the impulse with the tone OFF. Adjust R3 to the midpoint of that region over which the wave shape is smooth and the second peak is larger than the first.

3. Impulse Level. With the impulse attenuator switch in the 90 position and the vernier in the +10 position, adjust R5 so that the second peak has a zero to peak amplitude of 2.45 volts.

4. SFI scaling see previous procedure.

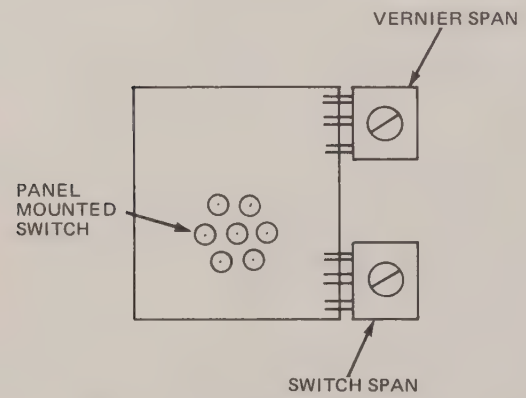


Figure 7 Attenuator Card Adjustments

e. PROCEDURE 5 – ATTENUATOR CARDS
(See Figure 7)

Three individual attenuators, which mount directly on the three attenuator switches, are provided. The range of the 10 dB vernier pot is adjusted by the Vernier Span adjustment. The magnitude of the 10 dB switch step is adjusted by the Switch Span adjustment. The procedure is as follows:

1. Set switch to full CW.

2. Set vernier to CW position (+10).

3. Back vernier off 10 dB and adjust Vernier Span for 10 db drop in output of appropriate quantity, i.e., Impulse, Noise, or Output Level as a result of this rotation. Note that 100 dBrn impulse has a 2.45 volt peak value, a 90 dBrn noise input has a .774 volt rms value, and a +10 dBm sine wave has a 2.45 volt rms value.

4. Return vernier to CW and switch attenuator down 10 dB. Adjust switch span for 10 dB drop in output.

5. Go back and forth to remove interaction.

The table below provides information useful in this procedure

dBm	dBRn	Voltage Across 600 ohms (volts)
+10	100	2.45
0	90	.774
-10	80	.245
-20	70	.0774
-30	60	.0245
-40	50	.0077
-50	40	.0024

As the lower levels background noise both in the 2A and in the test set up usually will cause the readings to be a dB or so high. Note that tone and noise levels are measured on an rms basis and Impulse is measured on a zero to peak basis, i.e., from ground to the highest (second) peak.

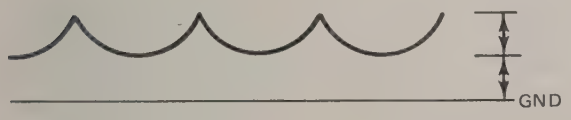
f. PROCEDURE 6 – SIGNAL PATH

This sequence involves the phase splitter, phase modulator and the amplitude modulator card of the 2A as well as the Frequency Translation section of the 2B. The procedure below assumes total maladjustment. Unless the adjustments have been meddled with this will never be the case for a unit in the field. Thus it is well to skip any steps in the procedure which are obviously preliminary such as "set pot xx to nominal midpoint etc." The adjustment complexity is required to make the PM and AM orthogonal, to minimize the PM, AM, and SFI at the translation frequency when frequency translation is on, and to make the AM symmetric. The first pass through the procedure is for qualitative results. The individual dial calibration procedures are given in paragraph 2 above.

1. Set R1, R2, and R3 of the Phase Splitter to nominal midpoint. Note R4 of Phase Splitter is set per paragraph 2 of this section. This is a non interactive adjustment.

2. Observe pin 9 of the phase modulation on the extender card or at the panel PM jack, with an oscilloscope. Set in 90° phase modulation at any convenient frequency. Sync the oscilloscope on the input or on the Modulation Oscillator (MOD jack) adjust R8 of phase modulator so that alternate cusps are equal as shown below.

3. With oscilloscope on same point, adjust R7 of phase modulator so that bottom of the cusps reach 50% of the peak level as shown below.



4. Remove the tone by setting the SIGNAL SOURCE the switch to EXT TONE position with nothing patched to that jack, the phase modulation remains at 90°. Using extender card, observe pin 16 of the phase modulator with an oscilloscope. Adjust R5 of phase modulator for symmetric amplitude modulation. Adjust R4 for zero dc average level.

5. Under the same conditions as 2 observe the voltage on pin 17. Adjust R2 for minimum output level and R1 for zero average level.

6. Using the SIGNAL SOURCE switch, connect the internal tone. Set tone for a frequency of 1000 Hz. Set phase hit dial to 180° and phase modulation dial to zero. Adjust R6 for minimum output at pin 16.

7. Proceed as in 5, but set phase hit dial to 0°. Adjust R3 for minimum output on pin 17.

8. Replace Phase Modulator card in socket and place the Amplifier Modulator card on the extender. Observe the OUTPUT jack with oscilloscope synced on TONE jack. Tone on, phase modulator on and to 90°. The modulation frequency is adjusted for "well-filled" jitter envelope on the CRT display. Adjust R83 of Amplitude modulator for minimum AM.

9. Set for 90° phase hits of 50 millisecond duration, fast rise time, at an interval of 0.1 second. PM remains on.

10. Adjust R1 of Phase Splitter, R3, R6, R7, and R8 of phase modulator, and R83 of the Amplitude Modulator for flat jitter (minimum AM).

11. Observe the voltage at pin 8 of the Amplitude Modulator (available on extender card). Turn PM off, Tone off, and AM on and to maximum. Adjust R121 and R123 for minimum output at ground level. Turn tone one and observe symmetric AM. R122 affects both the output level and the percent AM calibration. It is readjusted only if R124 does not have enough range to make the AM dial calibration correct.

FREQUENCY TRANSLATION

The following adjustments integrate the Frequency Translation into the 2A instrumentation.

1. Sync Oscilloscope on tone and observe test point 1 of 2B. With tone off, adjust R69 for minimum amplitude at translation frequency and R70 for zero dc average level. Turn tone on adjust R68 for symmetric positive and negative excursion.

2. Connect test point 2. with tone off adjust R56 for minimum amplitude at translation frequency and R57 for zero dc average level. Turn tone on and adjust R55 for symmetric positive and negative excursion.

3. Using Bradley Telcom PB-1 or PB-1C in Vector Notch mode (20 to 300 Hz) observe 2A output with Frequency Translation on 80% of full scale. Tone frequency is 1000 Hz. Adjust R56, R69, and R33 of 2B and R1 of Phase Splitter for minimum spot excursion.

4. Observe 2A output with oscilloscope synced on TONE jack. Adjust R55 and R68 for minimum AM and/or SFI on moving CRT pattern.

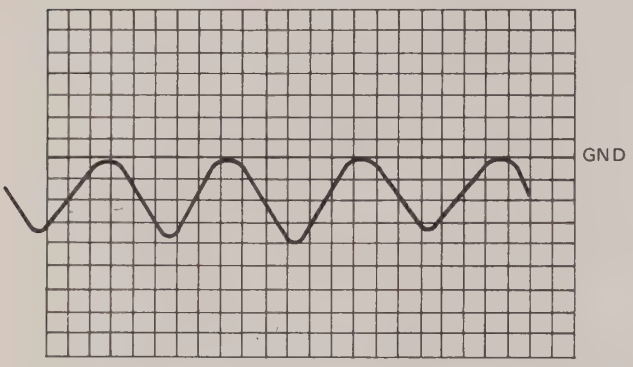
5. Set Frequency Translation to 20% of full scale and turn on PM to 90°. Adjust R43 for minimum wobble on moving jittered waveshape and R50 to make jitter crest straight line.

6. If necessary, adjust R57 or R70 to make jitter crest horizontal.

7. Remove PM and measure OUTPUT level with frequency translation on and off. Adjust R124 for identical reading within 1% (0.1dB).

SECOND HARMONIC

Observe Test Point 3 Figure 7 with an oscilloscope. Patch 1000 Hz, 600 ohm source, 0dBm sine wave to INPUT for HARMONIC DISTORTION. Adjust R86 and R87 for exact second harmonic sinewave. Adjust the amplitude with R98 approximately to 5V peak to peak. Adjust the amplitude with R88 for dc level such that second harmonic wave shape is tangent to ground potential, see below.



HARMONIC DISTORTION

Monitor 2B OUTPUT (-3dBm or -16dBm) with a PB-1 or PB-1C or spectrum analyzer. Set the second harmonic knob at 10% and the EVEN ORDER and NORMAL switch to the NORMAL position. Switch harmonic on and off. Make sure that the separation changes by 20 dB. This scaling is adjusted by R98. Upon completion, check that the voltage or TP3 is tangent to ground, see above.

THIRD HARMONIC

Scope on test point 4 same input as for second harmonic. Adjust R99, R100, and R119 for exact third harmonic sinewave. Adjust R101 for zero dc average level.

Monitor 2B output with PB-1 or PB-1C in third harmonic mode. With third harmonic knob set for 10% switch third harmonic on. Measure -20 dB spot motion between on and off. Adjust by R111. Throw third harmonic switch to the + and - positions. Make sure spot moves through the diameter of the -20 dB circle.

Adjust R129 for -3 dB 600 ohm loaded output level at the OUTPUT -3 dB jack.

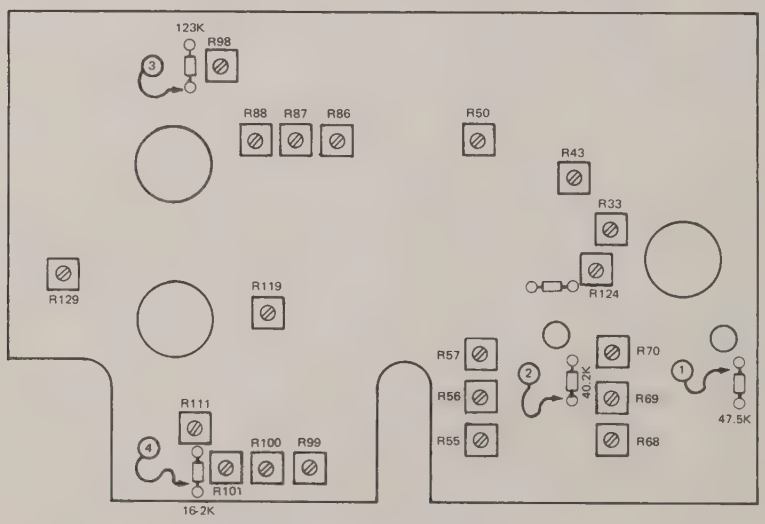


Figure 8 2B Adjustments

SECTION 3 – CLASSIFICATION OF LINE IMPAIRMENTS

1. PROPERTIES VERSUS IMPAIRMENTS

The linear steady state behavior characteristics of a transmission facility, for purposes of this discussion, are called the facility's PROPERTIES. These are described by the names:

Gain
Amplitude Distortion
Delay Distortion
Return Loss

The remaining transmission parameters are called the facility's IMPAIRMENTS. These are listed below in general terms each of which has several subclassifications.

Noise
Hits
Jitter
Distortion
Phase Intercept Distortion

PROPERTIES are not time-dependent. They can be completely characterized or described by a set of numbers or a plotted curve, e.g., gain versus frequency. As a result of adaptively equalized modems, the Line Properties, other than continuity (gain), have become less important as a cause of data problems. This is not true of IMPAIRMENTS. The behavior of impairments is complex and time variable so that they can neither be compensated for by circuitry, nor fully characterized by a set of numbers.

2. PROBING FACILITY FOR IMPAIRMENTS

In order to measure all impairments, except background noise, it is necessary to send a probing signal through a facility. For PCM facilities a probing signal is required even for noise measurements because quantizing noise, a major ingredient of PCM noise, is not present when there is no signal. Impairment behavior is monitored by observing the effects of the facility on the probing signal. This signal is usually a sine wave although band-limited noise, multiple tone signals, and harmonically enriched tones are also used. A sine wave is particularly easy to generate, as well as to monitor, and is therefore the basis for measurement of most impairments.

3. CLASSIFICATION OF LINE IMPAIRMENTS

The test tone, cos wt, as received via the communication channel, has disturbances which are synchronous with the test tone and disturbances which are not. This is reflected by the expression below in which g(t) and p(t) are respectively disturbances on the amplitude and phase of the received tone, and $H_k \cos (kwt + \Phi_{kw})$ is the harmonic distortion where k assumes integral values of 2 and over. N(t) are disturbances which are not correlated with, or synchronous with, the received tone.

$$\text{Received tone } V_r = A_w \left[1 + g(t)\right] \cos(wt + p(t)) + N(t) + H_k \cos (kwt + \phi_{kw})$$

Note that the reference in this discussion is the received tone $A_w \cos wt$. This takes no account of the channel phase shift, frequency shift, or echo. It does consider the difference in phase shift between the tone and its harmonics, Φ_{kw}, a matter of considerable significance, see below.

4. TONE-CORRELATED AMPLITUDE IMPAIRMENTS (See Figure 9)

The correlated disturbances which affect tone amplitude are incidental amplitude modulation, amplitude hits, amplitude steps, and dropouts. g(t) is that function of time which describes the amplitude variation of the test tone. It involves a relatively steady state component having a zero time average over the measurement bandwidth of the test instrument. This is "amplitude modulation".* g(t) also has a discontinuous component having a non-zero short time average. This component is called an amplitude hit or amplitude step. An amplitude step is a positive or negative step in g(t). It is measured with respect to the recent average level of the test tone subject to certain rise time and amplitude criteria. An amplitude hit is bi-directional, i.e., a step to a new value and then a return to more or less the original value. A dropout is a large decrease in level or short term loss of signal measured with respect to the received tone level at the time a test is started or other appropriate reference level. Note the difference between a dropout and a gain hit. The latter is defined with respect to a recent average level of the received tone, also there is no rise time criterion for a dropout, merely a level threshold and a minimum duration. The recently promulgated Bell System definition for a dropout is "a reduction in the received tone voltage level by more than 12 dB with respect to its value at the start of the test, which level stays below this threshold for 4 milliseconds or more". In suppressed phasor representation, amplitude jitter appears as a vertical line. Its length is a measure of the magnitude of the jitter. Hits appear as transient vertical excursions, either uni-directional or bi-directional dependent on their time history.

*Amplitude jitter is the sum of the effect of Incidental Amplitude Modulation and noise within the jitter filter bandwidth on the signal amplitude.

Figure 9 Tone Correlated Amplitude Disturbances

Figure 10 Tone Correlated Phase Disturbances

Figure 11 Uncorrelated Disturbances

		TONE-CORRELATED AMPLITUDE DISTURBANCES A		TONE-CORRELATED PHASE DISTURBANCES C		TONE-UNCORRELATED DISTURBANCES E	F	G
METHOD OF REPRESENTATION	TONE	AMPLITUDE MODULATION	GAIN HIT OR GAIN STEP	PHASE MODULATION	PHASE HIT OR PHASE STEP	SINGLE FREQ INTERFERENCE	NOISE	IMPULSE
MATHEMATICAL	$\cos \omega t$	$(1+K\cos gt)\cos\omega t$	$(1+H)\cos\omega t$	$\cos(\omega t + m\cos pt)$	$\cos(\omega t + N)$	$\cos\omega t + Q\cos mt$	$\cos\omega t + n(t)$	$\cos\omega t + I(t)$
TIME DOMAIN	AMPLITUDE, $T=\frac{\pi}{\omega}$, TIME	2K	H	2m	N	SFI ONLY TONE NOT SHOWN	NOISE ONLY TONE NOT SHOWN	IMPULSE ONLY TONE NOT SHOWN
FREQUENCY DOMAIN	AMPLITUDE, ωt, FREQUENCY	$\frac{K}{2}$, $\frac{K}{2}$, $(\omega-g)t$, ωt, $(\omega+g)t$	H, ωt	ωt, $(\omega+P)t$, $(\omega-2P)t$, $(\omega+2P)t$, J_2, J_1, J_1, J_2, $(\omega-P)t$	$\cos\omega t \cos N$, $\sin\omega t \sin N$	Q, ωt, $m t$	ωt	ωt
PHASOR	IN PHASE AXIS, ωt, QUADRATURE AXIS	2K, REF LENGTH	H	J_2, J_1, m	N	$(\omega-m)t$, Q, ωt		
SUPPRESSED PHASOR ONLY THE DISTURBANCE IN THE PHASOR PLANE IS DISPLAYED. THE REF TONE IS SUPPRESSED	ORIGIN	2K	EXCURSION IS TRANSIENT, H	2m	N			

5. TONE-CORRELATED PHASE IMPAIRMENTS (See Figure 10)

p(t) is that function of time which describes the short term phase variation. It involves a relatively steady state component having a zero time average over the instrumentation bandwidth (typically 20-300 Hz). This is called phase modulation.* Note in Figure 10 that the first sidebands of the modulated tone have an opposite sense as compared to the sidebands of an amplitude modulated tone, and that the second sidebands are AM-like sidebands. This can best be understood by considering the phasor representation. The phasor is rotated plus and minus with respect to reference. This produces a quadrature component between the reference tone and the received tone of amplitude sin Φ. It produces an amplitude variation of (1-cos Φ). The amplitude variation occurs at a double frequency; that is, as the phasor rotates from zero positively and back to zero, the amplitude goes through a complete cycle. And likewise, as the phasor rotates from zero negatively back to zero, the amplitude goes through a complete cycle. The complete mathematical expansion for a single frequency of phase jitter involves an infinite series of Bessel functions, of which the first sidebands correspond to J1 and the second sidebands correspond to J2. For the normally encountered phase modulation, the first two sidebands provide a sufficiently accurate description because of the low index of the modulation.

The Phase Hit column of Figure 10 represents before and after snapshots. It is necessary, in the frequency domain representation, to consider separately the sin wt and cost wt components of the frequency spectrum. Before the phase hit, the signal energy resides in the cos wt spectrum, and the sin wt spectrum is zero. The hit amounts to transfer of energy from the cosine to the sine spectrum. Note that for a true phase hit, the rms sum of the sine and cosine spectrums remains constant. There is no reference phase in a communication channel, thus there is no phase equivalent to a dropout. The test instrument tracks the phase of the received tone more or less slowly, and a phase hit is defined as an abrupt change with respect to the recent average phase of the received tone. Again, criteria are needed to define both the rate of change and the amplitude of the hit. Both steps (unidirectional abrupt changes in phase) and hits (phase discontinuities which are followed by a return to the starting phase) are found on typical common carrier facilities.

*Phase jitter is the sum of the effect of Incidental Phase Modulation and of noise within the jitter filter bandwidth on the signal phase.

In suppressed phasor representation phase modulation appears as a circle arc or as a horizontal line depending upon its frequency. If the display filter passes the second sideband the result will be a circle arc, if the modulation frequency is high enough so that the second sideband is outside the filter pass band, the display will be a horizontal line. Hits produce transient excursions which, depending upon hit transition time, are essentially circle chords aimed at the hit angle. The transients are eventually suppressed by the phase lock loop which, as does the AGC loop, corrects for a step transient in nominally 1/7 second.

6. NONSYNCHRONOUS OR UNCORRELATED IMPAIRMENTS (See Figure 11)

Nonsynchronous disturbances are single frequency interference (SFI), steady noise, and impulse noise. Quantizing, aliasing, and sampling interval noise in PCM systems is a special case of correlated noise which previously has been lumped with random noise. With the suppressed phasor technique such noise has a characteristic pattern. Figure 11 shows a single frequency interfering tone, which, when added to the test tone, produces a phasor domain signal as shown. There is a unit length vector directly upward representing cos wt, plus the additive SFI vector spinning around the tip of the unit vector. The amplitude of this spinning vector is the amplitude of the single frequency interfering tone relative to the tone amplitude, and the circular frequency of its rotation is the difference frequency between the test tone cost wt and the single frequency interfering tone. It is, in effect, a single sideband modulation.

Steady noise is similar to single frequency interference except that it comprises a spectral distribution of varying amplitude interfering signals. As also depicted in Figure 11, a phasor representation, there is no single, unique spinning additive vector. The instantaneous value of the output can be any place within the relatively jagged circle. The amplitude of the "circle" is a measure of the summation of the noise factors as they reinforce and interfere with each other.

Figure 11 also shows an impulse. Impulses are highly variable and usually brief, lasting only a few milliseconds. A typical signal to steady noise objective is 24 dB. A comparable signal to impulse noise objective is 5 dB. The difference is acceptable partly because of the short duration of impulse noise, and partly because hit ratios are based on rms/peak, whereas steady ratios are rms/rms. Figure 11 represents the instantaneous situation during the impulse which will typically have its energy in some band that is not necessarily symmetric about the tone, and which will not necessarily define a circle in the phasor diagram.

1. PHASE JITTER

Phase Jitter Test Sets typically determine the variation in zero crossing of a test tone either with respect to a reference tone phase locked to the received tone, or with respect to successive zero crossings. If a phase-locked reference is used it has a long time constant so as not to track the jitter. The test tone is normally 1004 or 1020 Hz, and the bandwidth is typically 20 to 300 Hz; that is, the test instrumentation is sensitive to phase modulation of the test tone in the 20 to 300 cycle bandwidth. Either instrumentation is incapable of distinguishing between true phase modulation and uncorrelated noise which occupies the same frequency interval as the jitter, i.e., 700 to 980 Hz and 1020 to 1300 Hz for a 1000 Hz test tone. The 1020 Hz test tone is used with "T" carrier to distribute the quantizing noise foldback products over the spectrum rather than having it concentrated at a few frequencies as happens when the tone and the sampling frequency (8KC for T carrier) are harmonically related.

One jitter set attempts to resolve noise and jitter for T carrier with a variable bandwidth filter, and by accommodating a wide range of tone frequencies. The theory is that by manipulating these variables (filter bandwidth and tone frequency), the relative magnitude of jitter and quantizing noise can be estimated. Another manufacturer's jitter set includes a spectrum analyzer, ostensibly to separate the noise from the phase jitter. Since noise in the 700 to 1300 Hz band produces the same spectral energy distribution as true phase modulation around the carrier in the 0 to 300 Hz band, such separation involves shifting the tone frequency and/or level to see which spectral components are tone dependent. A simpler approach is used on non-compandored facilities. The jitter is measured at two different tone levels. Background noise, because it tends to remain constant, makes a larger relative contribution to jitter at lower tone levels than does the true phase jitter, whose energy level varies with the tone. Thus, if the jitter reading remains relatively constant as the tone level is changed, it is indicative of true phase modulation as the predominant disturbance. If the jitter reading is inversely proportional to tone amplitude, it is indicative of (tone independent) noise as the source of apparent jitter.

2. AMPLITUDE JITTER

Amplitude Jitter exists in communications networks and, as combination phase/amplitude encoded and/or vestigial sideband modems come into greater use, one may expect a similar experience to that of phase jitter a few years ago. Indeed, the phase jitter problem in the normal (non-trouble situation) network is now under control while the effort has not yet begun with amplitude jitter. The Bradley Telcom Model 75LMA, measures both phase jitter and amplitude jitter in the 20 to 300 Hz bandwidth. This instrument uses the same post detection filter and peak detector for either PM or AM measurement.

As a diagnostic tool in combination with phase jitter, amplitude jitter is perhaps better than the diagnostic techniques for separating noise and jitter which are described above because it can be used at fixed tone level and tone frequency, i.e., someone does not have to camp at the far end. Uncorrelated noise makes an equal contribution to amplitude jitter and to phase jitter. Thus, widely different amplitude and phase jitter readings (reduced to an equivalent sideband energy basis) are an unequivocal indication that steady noise (including quantizing noise) is not the offender. If the amplitude and phase jitter readings are equivalent, then either the amplitude jitter and phase jitter are equal (unlikely, since they come from different sources both of which must be in trouble), or the readings are the result of uncorrelated noise (steady noise or SFI). One significant advantage of this is that this diagnostic technique is independent of the type of facility (compandored or not). This makes it convenient for the common carrier craftsman; he does not have to research the channel configuration. It also makes the technique feasible for the customer, who may not have access to channel configuration information.

Of course, communication channel problems are quite diverse. It is possible to devise instrumentations and measurement techniques to identify individual problems, but the existence of other impairments, which confuse the results, is virtually invariable. The experience of the craftsman is what makes any diagnostic system effective.

3. STEADY NOISE

Steady noise is measured

1. With or without a holding tone
2. Relative to the signal level or in terms of its own absolute level
3. With a variety of filter weightings
4. With an rms or quasi-rms detector

1. With a holding tone on a compandored facility, i.e. a facility with agc, a holding tone is required to establish the gains of all the agc amplifiers at some nominal level rather than having them wide open, thus emphasizing noise. In this case, steady noise is dependent upon the distribution of agc gains in the channel so that the more meaningful parameter is signal to noise ratio. The tone is notched out by filtering, its amplitude measured, and the energy in the signal minus the tone is measured. The ratio of these two quantities is displayed as signal to noise ratio. Notch noise includes steady noise, phase jitter sideband energy, amplitude jitter sideband energy, and harmonic distortion, i.e., everything but the tone fundamental.

2. When noise is measured without a holding tone on non-compandored facilities, a quiet termination is patched to the source end, and the noise level read at the receiving end. No notching or signal manipulation is involved other than detection, and the noise level is typically displayed in dBrnc. Zero level dBrnc = -90dBm.

3. Filters are used to weight the noise components. 3Kc flat, C-message, and sophometric weightings are commonly used. C-message is an experimentally derived weighting which represents the annoyance and reduced intelligibility factors of noise at various frequencies on a standard telephone instrument with reference to a 1000 cycle tone. C-message weighting was established in the United States; sophometric weighting is a similarly determined form of filter weighting which was established in Europe. Both of these have a low frequency roll-off resulting in a high attenuation at 60 Hz (and 50 Hz) relative to the 3Kc flat filter. These filters are all quite flat in the bandwidth used for data transmission, so that C-message weighting (or sophometric) is also used in data channels. Diagnostically, a high noise reading with a 3Kc flat filter relative to the reading with either of the others is an indication of the presence of power line or ringing tone.

4. Detectors. Noise level is the rms power level of the noise signal, and is properly measured by a true rms detector. It is also frequently measured by a "quasi rms detector." A quasi rms detector is a combination of the outputs of an average value and a peak value detector, with the contributions of each weighted so that the output is correct with either a pure sine wave or for a Gaussian distribution of the input signal (but not for most energy distributions other than these two). An average value detector normally consists of a conventional averager scaled so that it reads true rms value for a sinusoidal input. The average value-calibrated rms detector understates the rms value of a Gaussian distribution input by 1dB.

4. HIT COUNTERS

The definition of hits is complicated by the need to include a time history description of the event and of the measurement interval (typically 15 minutes) over which the count is accumulated.

Also, typically, hit counters have a built-in deadband of 1/7 sec.; i.e., after a hit event has been identified, it is counted in the one or several applicable registers, and the detection circuitry is disabled for 1/7 second. This procedure came about historically; early hit counters used electro-mechanical counters whose response time was limited. Not only was a great deal of data developed under these conditions, but criteria were established for relating this data to line performance on a continuous basis.

In the case of a dropout, after the event is counted, all hit detection circuits are disabled for the duration of the dropout. They are not again enabled until the test tone exceeds the threshold level for a specified period, i.e., 4 milliseconds.

5. IMPULSE

Impulses are measured on a peak power basis with essentially negligible duration requirements, e.g., if the peak power exceeds the threshold level for 50 microseconds, an impulse is counted. Many impulse counters have multiple threshold level detectors counting into separate registers. Logic of these instruments is such that, for a given impulse, a threshold event is counted in all registers up to the highest register which is triggered. The results displayed by the registers represent a statistical distribution of impulse amplitudes.

6. PHASE HITS

Phase hits are time coherent; i.e., synchronous with the test tone, although short duration hits of this type may not appear to be so because they start at a random time during the tone cycle. The problem of distinguishing between these correlated hits and impulse hits, which are uncorrelated, is handled by establishing a "guard band" of two to four milliseconds (depending upon particular instrumentation). This is long enough to test the signal for coherency (and quadrature). The four millisecond interval (of the older instruments) was established also because 90% of measured impulse hits are shorter in duration than four milliseconds; hence the confusion between impulse and phase hits is reduced. There are subsidiary difficulties associated with filter ringing and/or delay which reduce this apparent time margin. In order not to miss short phase hits, a shorter guard band is preferable *if* impulse hits can be discriminated against.

Another problem in measuring a phase hit is indicated in Figure 12, which plots phase vs time for a hit. The hit is not instantaneous, channel bandwidth and other considerations assure that. The phase reference, as generated by the instrument phase lock loop, tracks the test tone phase more or less slowly. Thus a higher hit threshold is reached than the hit counter threshold setting, before a count is recorded. Of course, the ideal hit counter matches the tracking rate of the modem for which the channel is to be used, but this is impractical for a general purpose instrument.

There does not yet exist a generally accepted definition of instrument response to non-instantaneous, i.e., all, phase hits and, as a result, phase hits counted by different manufacturers' hit counters are not the same (because most phase hits have a rise time from two to five milliseconds).

7. GAIN

The considerations for gain hits are essentially the same as for phase hits except that test tone amplitude rather than test tone phase is examined.

8. DROPOUT

This is an absolute level threshold which is established at the start of a test. There are no rise time considerations; a dropout is counted if the level falls below this threshold for a specified time. A typical definition is 12dB below start of test for at least 10 milliseconds. There is no upper limit on the dropout time, and only one dropout is counted no matter how long it lasts, until the signal goes above the threshold for at least 4 milliseconds, whereupon the dropout is considered over. The next excursion below threshold (for at least 4 milliseconds) is counted as another dropout.

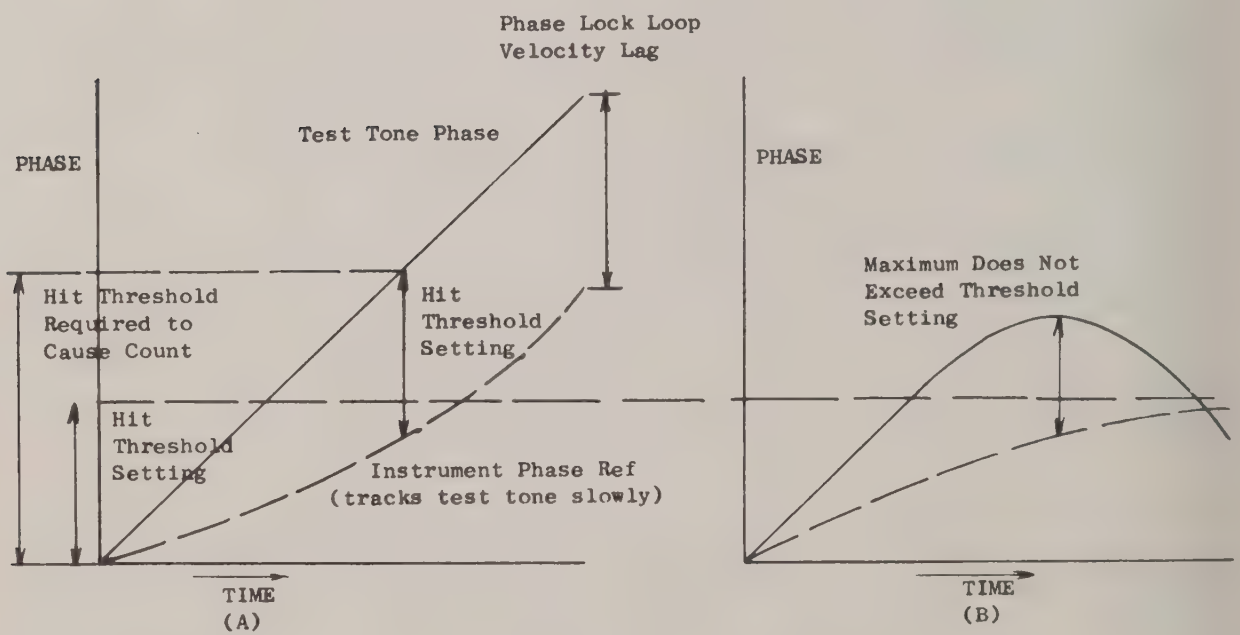


Figure 12 Effect of Hit Dynamics on Hit Count

SECTION 5 – MISCELLANEOUS APPLICATION COMMENTS

1. FAMILIARIZATION PROCEDURE

To stop the internally generated tone on an oscilloscope display for viewing phase jitter, sync on the tone oscillator of the synthesizer. One can also view AM and SFI (Single Frequency Interference) in this way. In this case the tone component will be stationary and the modulation will be moving. It is also possible to sync from the modulation oscillator so that the envelope is stationary and the tone moves. It is also possible to sync on the HIT so that hit events are stationary.

1. Scope Sync External, sync off the TONE jack. Vertical input on the OUTPUT jack. The synthesizer has all impairment switches off, SIGNAL SOURCE (center left) switch in the TONE FREQUENCY position TONE FREQUENCY (lower left) dial at 1000 (Hz). OUTPUT LEVEL 0dbm (switch -10, pot +10).

Set scope sweep rate for two or three cycles of the tone visible and the vertical gain so that the sine wave is on scale.

2. "EXT TONE (input for modem or other signal source). Switch SIGNAL SOURCE back and forth and observe the sine wave disappears. If modem signal were present, this would appear at the output in the "EXT TONE position.

3. MODULATION FREQUENCY to 150 Hz (or thereabouts) x 1 on switch and 150 on vernier.

4. PHASE MODULATION ON, move PHASE MODULATION DIAL up to cause jitter.

5. Vary modulation frequency to observe that capability.

6. PHASE MODULATION OFF, AMPLITUDE MODULATION ON and repeat.

7. AMPLITUDE MODULATION off and SFI on and repeat. SFI is additive tone at modulation frequency. SFI appears in the output as an additive tone, not as modulation. Hence it is subject to the frequency response characteristics of the output transformer and, below 60Hz is attenuated.

8. SFI off, NOISE on and vary level.

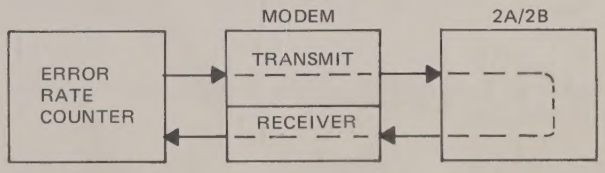
9. Switch Tone off (SIGNAL SOURCE) to observe noise by itself.

10. Also observe EXT inputs. Set the NOISE Level to a high value but leave the NOISE ON/OFF switch in the off position. Patch NOISE output jack successively to EXT PM, EXT AM and EXT ADD (with tone on). Observe pm (movement at zero crossings) and am (movement at peaks).

11. HITS either manual by PUSH TO HIT button (toggle up) or continuous (toggle down). The interval, rise time, and duration of the hits are adjustable. Note that the duration must be greater than the rise time and the interval must be greater than the duration plus the rise time. A good demonstration is 0.1 second interval, 50 millisecond duration, 0.2 millisecond rise time. Nothing happens until the Phase Hits Toggle, Gain Hits Toggle or Impulse toggle is turned on, even though the hit pulse is being generated. Turn phase hits on and adjust hit amplitudes up and down, ditto gain hits. The gain hit dial includes dropouts.

Impulse - fixed waves shape which occurs only on leading edge of hit. Observe it with and without tone (switch to -80, pot +10).

2. **MODEM EVALUATION CAPABILITY** - Set up, with all impairments off, so that error rate counter, see below,



is not counting any errors. This usually involves only getting the levels and impedances correct, however, it may be more complicated if the modem is of the manual equalization type.

Error rate counters generate bit sequences which are sent through a line (synthesizer). The received data stream is compared with the transmitted data stream and either, or both, bit errors and block errors are counted. A block is a group of bits of arbitrary but specified length. A block error is counted when one or more bit errors occurs in a given block.

The characteristics not simulated in the 2A/2B are the steady state impairments, i.e., gain vs. frequency and envelope delay vs frequency. The 2A Synthesizer is essentially equivalent to a Bell System C5 line. This specifies the steady state parameters above. Actually it is better in gain vs. frequency and slightly over on the C5 line spec for envelope delay vs. frequency. See data below.

3. TYPICAL DATA

Frequency Characteristics of 2A or off 2A and 2B with Frequency Translation Off

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	–.1	579.
437.	.0	313.
594.	.0	203.
750.	.0	145.
906.	.0	105.
1062.	.0	76.
1219.	.0	54.
1375.	.0	37.
1531.	–.1	22.
1687.	–.1	11.
1844.	–.1	.
2000.	–.1	–11.
2156.	.0	–22.
2312.	.0	–32.
2469.	.1	–41.
2625.	.1	–47.
2781.	–.1	–49.
2937.	.0	–42.
3094.	–.1	–31.
3250.	.0	–24.

Frequency Characteristics of 2A and 2B with Frequency Translation On

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	.0	1169.
437.	.0	635.
594.	.0	413.
750.	.0	293.
906.	.0	214.
1062.	.0	157.
1219.	.0	112.
1375.	–.1	77.
1531.	–.1	47.
1687.	–.1	22.
1844.	–.1	.
2000.	–.1	–21.
2156.	–.1	–40.
2312.	.0	–58.
2469.	.0	–73.
2625.	.0	–85.
2781.	.0	–92.
2937.	–.1	–90.
3094.	–.1	–83.
3250.	.0	–82.

Frequency Characteristics of Harmonic Distortion Section

FREQ (HZ)	LOSS (dB)	EDD (MICROSEC)
281.	–.2	–6.
437.	–.1	–4.
594.	–.1	–4.
750.	.0	–5.
906.	.0	–5.
1062.	.0	–5.
1219.	.0	–4.
1375.	.0	–4.
1531.	.0	–2.
1687.	.0	–1.
1844.	.0	.
2000.	.0	.
2156.	.1	.
2312	.1	–3.
2469	.2	–4.
2625.	.2	–5.
2781.	.2	–4.
2937.	.2	1.
3094.	.1	10.
3250.	.2	17.

Modems are equalized for line characteristics, particularly envelope delay. Envelope delay has the effect of making the high frequency components of the modem signal travel or arrive at a different time from those low frequency components which were launched into the line at the same time. Some high speed modems are equalized manually, mostly the older designs, most are equalized automatically.

When connecting a modem note that the peak to peak input level at the EXT INPUT jack is 2.2 volts maximum. This corresponds to 0dbm for sine wave but not modem signals. Modems normally have higher crest factors. Note the gain through the synthesizer is 0db (unity) when the OUTPUT LEVEL dial is set to 0dB (-10 switch, +10 pot). Frequently a loss of 16db is desired in the transmission path or rather an output level is desired of -16dB. If this is what is wanted with an input signal coming in, for example, at -3, then the OUTPUT LEVEL switch should be set to -13 to give a minus 16dBm net output level.

If a scope is kept on the output at all times it is possible to switch back and forth between the modem and the internal tone so that the impairment can be viewed as it effects the tone and hence recognized by inspection.

The modem is now playing a data stream via the synthesizer, out of, and back into the bit error rate counter. All impairments are "off" and the error rate counter is counting no errors. This will involve matching levels at the synthesizer input and output as well as matching impedances. Make sure that the peak to peak amplitude of the input signal to the synthesizer is 2.2 volts or less. This may involve transmitting at some level between -3 and 8dB, depending upon the cross factor of the modem. The output level of the synthesizer can be adjusted relative to the input level. The synthesizer has zero db gain or unity gain at zero dbm output level setting. To increase the level through the synthesizer, set it above 0dB, up to +10dB (which provides 10dB gain through the 2A). To drop the level set it below 0dBm by the attenuation desired.

4. PHASE MODULATION

Set the modulation frequency to, for example 60 Hz, switch the toggle switch on and move the phase modulation level slowly up until the error rate counter starts counting errors. Log this magnitude of jitter.

Repeat at other frequencies which may be of interest, typically, 20 to 300 Hz. The result of this is a curve which defines the sensitivity of the modem to phase jitter frequency.

5. AMPLITUDE MODULATION

Same as above except with amplitude modulation dial. Figure 13 relates % AM to amount of AM sideband energy.

6. SINGLE FREQUENCY INTERFERENCE (SFI)

This is complicated because the dial is in per cent relative to zero level signal (at output) - *when working from the external tone input only*, see correction factor tables below. For internal tone the dial is calibrated correctly at all output levels.

EXTERNAL TONE LEVEL AT 2A SYNTHESIZER OUTPUT JACK, dBm	SFI SCALE MULTIPLIER
0	x1
–3	x1.41
–6	x2
–9	x2.83
–12	x4
–16	x6.31
–20	x10

The SFI frequency, which is most disturbing to modems is in the vicinity of modem carrier frequency (typically from 1600 to 2200 Hz). Within above ground rules one can generate the same kind of curve for SFI. The most commonly encountered SFI tone in actual operation in the power line frequency. Figure 14 relates % SFI to dB SFI.

7. NOISE

Noise level is in dBrn, weighting is flat to 3.5KHz with 12db per octave roll-off thereafter. dBrn is dbm -90, i.e., 0dBm = -90dBrn. The critical factor is the noise level in relation to the tone level (signal to noise ratio).

For example, a 20dB signal to noise ratio means that the noise level itself is 20dB below the tone level, i.e., if the tone level is -16dBm the noise level is -36dBm or +54dBrn (dial setting of noise attenuator).

The situation is complicated by the noise source bandwidth and/or any filtering that may be in the line for simulation purposes. Completely consistent results between simulation methods requires that the noise source have identical bandwidths or are measured in such a way that the measurement response is equivalent for the two sources.

8. HARMONIC CALIBRATION

The harmonic dials of the 2B are calibrated in % (Relative). This can be converted to dB (Relative) by referring to Figure 14.

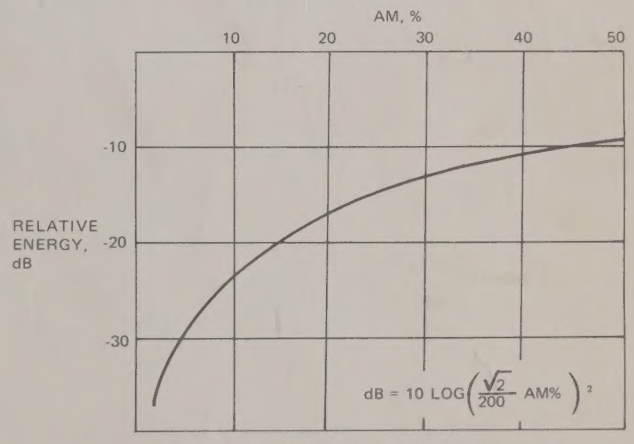


Figure 13 % AM vs Relative AM Sideband Energy

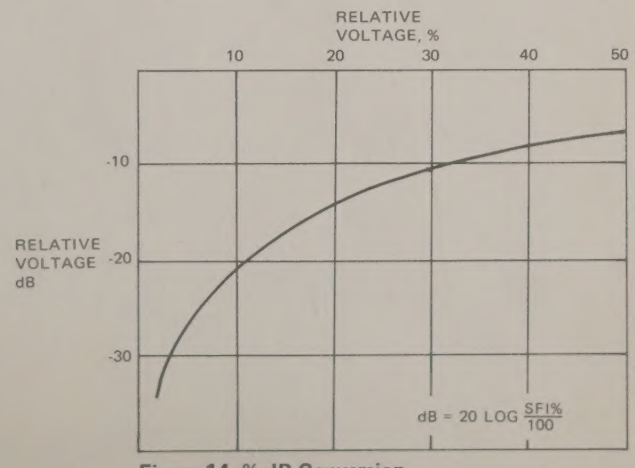
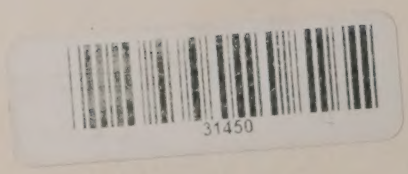


Figure 14 % dB Conversion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